# REMARKS

In the patent application, claims 1-7, 9-17, 19-34 and 36 are pending. In the final office action, all pending claims are rejected.

## A. 102 Rejection

At section 2 of the office action, claims 1-7, 9-17 and 19-34 and 36 are rejected under 35 U.S.C. 102(e) as being anticipated by *Ravi et al.* (U.S. Patent No. 6,292,834, hereafter referred to as *Ravi*) and *Wang et al.* (U.S. Patent No. 5903,673, hereafter referred to as *Wang*) incorporated by reference.

The Examiner states that *Ravi* teaches a method for multimedia streaming as claimed in claim 1.

### A.1 Claim Limitations in Claim 1

It is respectfully submitted that claim 1 includes the limitations of

1) defining in a client in a multimedia streaming network at least one parameter for determining a rate adaptation operating range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, and wherein the rate adaption operating range is used for rate adaptation between the server and the client;

2) providing to the server information indicative of said at least one parameter;

3) adapting in the server the data amount to a reception rate at the client based on said at least one parameter, and

4) adjusting in the client packet transfer delay variation based on said adapting, wherein the one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

A.2 Rejection of Claim 1

In rejecting claim 1, the Examiner points to column 6, lines 33-47; column 7, lines 16-25 and column 8, lines 26-45 to show that *Ravi* discloses adapting in the server the data amount to a reception rate at the client based on said at least one parameter. The Examiner further points to col.10, lines 1-67 of *Wang* to show that *Ravi*, by incorporation by reference, discloses transmitting a frame based on network condition, and adjusting the cumulative bandwidth balance amount of bandwidth time consumed by current frame. In particular, the second loop rate control 204 adds to the cumulative bandwidth and the current frame and subtract from the cumulative bandwidth balance the amount of bandwidth time consumed by the current frame. Thus, *Ravi* discloses the difference between a sampling time and transmission time of a packet at the server.

A.2.1 Previous Argument Against Rejection of Claim 1

The rejection of claim 1 in the final office action is identical to the rejection in the non-final office action, mailed June 8, 2009.

In response to the non-final office action, applicant points out that *Wang* discloses using a Q adjuster 116 to adjust the Q 114 based on the output of the primary open loop rate control 202 and the output of the secondary closed loop rate control 204 (Figure 2; col.9, lines 33-43). The quantization parameter Q used in encoding will affect the amount of encoded data of a frame and, therefore, the cumulative data amount to be transmitted. However, adjusting the cumulative data amount to be transmitted is different from changing the sampling time. A sampling time is when the packet is started to be sampled relative to when the packet is started to be transmitted, for example. The sampling time can be moved up or delayed relative to the transmission time without changing the encoding resolution. Likewise, the cumulative data amount of the packet can be decreased by changing the quantization parameter Q without changing the sampling time.

Therefore, *Ravi* and *Wang* fails to disclose adjusting a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

A.2.2 Examiner's Response to Applicant's Previous Argument

At section 5 of the final office action, the Examiner states that "According to the present application, before transmittal of a stream, the data needs to be encoded by the server. The rate

of encoding is called a sampling rate" (page 8, lines 4-6); "Wang, as incorporated by *Ravi*, teaches the server adjusts the value of Quantization (Q) to encode the stream. This value of Q is adjusted based on a bandwidth level known by the server. *Ravi* teaches this bandwidth value is provided by the client. Adjusting the Q teaches adjusting the **sampling rate** of the streaming data" (page 8, lines 8-12, emphasis added)

A.2.3 Sampling Rate v. Sampling Time

It is respectfully submitted that **sampling rate** is not the same as **sampling time**. Sampling rate is the frequency at which the data is sampled as measured by bytes/second, for example. Sampling time is when the part of the streaming data is sampled. The difference between sampling time and transmission time is measured by seconds, for example. **Sampling rate** is not part of the claimed limitation.

The Examiner fails to show that *Ravi* and *Wang* disclose adjusting a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

A.2.4 *Wang* Fails to Disclose Adjusting Difference Between Sampling Time and Transmission Time

It is respectfully submitted that, in col.10, lines 21-59, *Wang* discloses:

> *In test step 410, secondary closed loop rate control 204 determines whether the cumulative bandwidth balance is greater than the upper threshold of the range determined in step 404. If the cumulative bandwidth balance is within the desired range, processing transfers to test step 414 which is described more completely below. Conversely, if the cumulative bandwidth balance is greater than the desired range, excess bandwidth is accumulating and processing transfers to step 412 in which secondary closed loop rate control 204 decreases Q 114. Accordingly, video image quality is increased at the expense of increased bandwidth consumed by subsequent frames. This is appropriate since unused accumulating bandwidth is detected and using such bandwidth improves the overall perceived quality of the motion video image. In one embodiment, Q 114 is adjusted 1% for every 3% of the upper threshold that is exceeded by the*

*cumulative bandwidth buffer. After step 412, processing of the current frame by secondary closed loop rate control 204 completes.*

*In test step 414, secondary closed loop rate control 204 determines whether the cumulative bandwidth balance is less than the lower threshold of the desired range determined in step 404. If the cumulative bandwidth is within the desired range, processing of the current frame by secondary closed loop rate control 204 completes. Conversely, if the cumulative bandwidth balance is below the desired range, bandwidth is being consumed at too great a rate and processing transfers to step 416 in which secondary closed loop rate control 204 increases Q 114. Accordingly, image quality is sacrificed to conserve bandwidth used by subsequent frames. Therefore, small excesses in consumed bandwidth which are undetected by primary open loop rate control 202 but which accumulate over time are detected by secondary closed loop rate control 204 and available bandwidth is not exceeded. In one embodiment, Q 114 is adjusted 1% for every 3% of the lower threshold that exceeds the cumulative bandwidth buffer. After step 416, processing of the current frame by secondary closed loop rate control 204 completes.*

The quantization parameter Q used in encoding will affect the amount of encoded data of a frame and, therefore, the cumulative data amount to be transmitted. However, adjusting the cumulative data amount to be transmitted is different from changing the sampling time. A sampling time is when the packet is started to be sampled relative to when the packet is started to be transmitted, for example. The sampling time can be moved up or delayed relative to the transmission time without changing the encoding resolution. Likewise, the cumulative data amount of the packet can be decreased by changing the quantization parameter Q <u>without changing the sampling time.</u>

<u>A.2.5 *Ravi and Wang* Fails to Anticipate Claim 1</u>

*Ravi* and *Wang* fails to disclose adjusting a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server. *Ravi* and *Wang* fails to disclose adapting in the server the data amount to a reception rate at the client based on said at least one parameter, and adjusting in the client packet transfer delay variation based on said

adapting, wherein the one parameter comprises a shift amount in time indicative of a difference between a sampling time and a transmission time of a packet at the server.

For the above reasons, *Ravi* and *Wang* fails to anticipate claim 1.

A.3 Rejection of Claims 2 and 3

In rejecting claims 2 and 3, the Examiner further states that *Ravi* discloses a minimum shift amount (decrease bandwidth threshold 512, column 7, lines 35-45; *Wang*, col. 10, lines 1-20) and a target shift amount (delta playtime and shift amount, column 8, lines 36-65).

Applicant respectfully disagrees.

A.3.1 *Ravi and Wang* Fails to Disclose Claim Limitations in Claims 2 and 3

At column 7, lines 16 to 25, *Ravi* discloses:

> *FIGS. 5A, 5B, 5C, 5D and 5E, are detailed flowcharts illustrating steps 410, 420, 430, 440 and 450, respectively, of FIG. 4. In step 410, the performance variables are computed. Next, in step 420, the computed performance variables are used to determine if it is desirable to decrease the bandwidth, and if so, then in step 430, the bandwidth is decreased. If a bandwidth decrease is not desirable, then in step 440, the performance variables are used to determine if it is desirable to increase the bandwidth. If a bandwidth increase is desirable, then in step 450, the bandwidth is increased.*

In the above paragraph, *Ravi* discloses the client computer 240 computes the performance variable (step 410), including computing playtime and delta_playtime (step 513); decreasing the bandwidth (step 430) and sending a reduce_bandwidth message to the server (step 537); or increasing the bandwidth (step 450) and sending an increase_bandwidth message to the server (step 522). According to *Ravi*, the term "bandwidth" is synonymous to the "transmission rate" (column 6, line 63 to column 7, line 5).

At column 7, lines 35-45, *Ravi* discloses:

*FIGS. 6A and 6B are two halves of a flowchart illustrating the dynamic determination of the Upper INC_BW threshold and the DEC_BW threshold, step 512 in greater detail. In step 612, the difference (D1) between the Current_Time and the previous time the dynamic bandwidth selection method was invoked is computed. In step 614, the difference (D2) between the timestamp of the last data packet currently in playout buffer 366 and the timestamp of the last data packet in playout buffer 366 during the previous invocation of the Adjust_Bandwidth procedure, is computed. In step 616, the difference (D3) between the number of bytes received by the previous invocation and the number of bytes currently received (by the current invocation) is computed.*

As pointed out above, *Wang* only discloses adjusting the quantization parameter Q 114 in order to adjust the cumulative data amount to be transmitted. *Ravi* only discloses adjusting the transmission rate. Thus, *Ravi* and *Wang* does not disclose or suggest adjusting the difference between the sampling time and the transmission time.

At column 8, lines 26 to 65, *Ravi* discloses:

*FIG. 7A is a flowchart illustrating the computation of variables Playtime and Delta_Playtime, step 513, in greater detail. In step 710, Playtime is set to the Duetime of the last packet in playout buffer 366. The computation of the Duetime is described in greater detail in step 740 below. Client computer 240 determines the change in the Playout_Buffer_Size (step 720). The Delta_Playtime is set to the difference between the current Playtime and the Playtime at the previous invocation of the Adjust_Bandwidth procedure (step 730). Variables Playtime and Delta Playtime provide exemplary absolute and relative measures, respectively, of the Playout_Buffer_Size, the number of data packet(s) in playout buffer 366.*

*FIG. 7B illustrate the determination of the Duetime of a data packet (step 710). First, the Base_TS is set to the timestamp of the first packet received by client computer 240 (step 712). The Base_Time is set to the time when the first packet was received (step 716). The TS is set the timestamp of the data packet of interest (step 746).*

In the above paragraphs, *Ravi* only discloses how the client computer 240 computes the Playtime and Delta Playtime (step 513, Figures 5a and 7a)

A.3.2 *Ravi and Wang* Fails to Anticipate Claims 2 and 3

None of the cited passages in *Ravi* and *Wang* discloses adjusting the difference between the sampling time and the transmission time. *Ravi* and *Wang* fail to disclose that the shift amount is substantially equal to or greater than the difference. Thus, *Ravi* and *Wang* fail to anticipate claims 2 and 3.

Furthermore, claims 2 and 3 are dependent from claim 1 and include further limitations. For reasons regarding claim 1 above, claims 2 and 3 are distinguishable over the cited *Ravi* and *Wang* references.

A.4 *Ravi and Wang* Fails to Anticipate Claims 4, 5 and 9

As for claims 4, 5 and 9, they are dependent from claim 1 and include further limitations. For reasons regarding claim 1 above, claims 4, 5 and 9 are also distinguishable over the cited *Ravi* and *Wang* references.

A.5 *Ravi and Wang* Fails to Anticipate Claims 11-15, 17, 19-24 and 26-31

For reasons regarding claim 1 above, *Ravi* also fails to anticipate independent claims 11, 21 and 26, and dependent claims 12-15, 17, 19, 20, 22-24 and 27-31.

B. 103 Rejection

At section 4, claims 6-7, 10, 16, 25 and 32-34 are rejected under 35 U.S.C. 103(a) as being unpatentable over *Ravi*, in view of *Nilsson et al.* (U.S. Patent Application Publication No. 2005/0172028, hereafter referred to as *Nilsson*).

B.1 Rejection of Claim 6

In rejecting claim 6, the Examiner states that *Ravi* fails to disclose that the parameter comprises a shift amount in time indicative clock shift between the server and the client, but points to *Nilsson* for disclosing such feature (paragraphs [0133]-[0134]).

B.2 Claim Limitations in Claim 6

It is respectfully submitted that claim 6 includes the limitations of:

1) defining in a client in a multimedia streaming network at least one parameter for determining a rate adaptation operating range, wherein the streaming network comprises a server configured for providing streaming data to the client, the client having a receiver buffer for storing at least part of the streaming data to compensate for a difference between data transmission amount by the server and usage amount of the streaming data by the client so as to allow the client to have sufficient amount of streaming data to play out in a non-disruptive manner, and wherein the rate adaption operating range is used for rate adaptation between the server and the client;

2) providing to the server information indicative of said at least one parameter;

3) adapting in the server the data amount to a reception rate at the client based on said at least one parameter; and

4) adjusting in the client packet transfer delay variation based on said adapting, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client

B.3 *Nilsson* Fails to Disclose Client Sending to Server a Parameter Indicative of Clock Shift Amount

In paragraph [0133], *Nilsson* discloses:

> *To determine the exact amount of data in the client's decoding buffer 41, the server also needs to know the timestamp of the data packet that the client is currently decoding and presenting. The server 10 calculates this using two assumptions: firstly that the client 40 starts decoding immediately after the server 10 sends the first packet; and secondly, that the client's clock does not drift significantly from the server's clock in the duration of streaming.*

In paragraph [0134], *Nilsson* discloses:

*In practice both assumptions have been found to be valid. The client 40 is designed to start decoding immediately on receipt of data, and so any error on the server's estimated presentation time would result in an underestimate for the amount of data in the decoding buffer 41, which as explained above is not a problem. Drift between the client's and server's clocks during a typical streaming session is most likely to be negligible compared to the amounts of data being buffered. For example, with a difference of 100 parts per million, it would take 10000 seconds, or nearly three hours, for a drift of one second to occur. In the rare case of a large amount of drift accumulating, the client 40 can warn the server 10 by use of a control message, such as the one described earlier that is sent for decoding buffer underflow.*

It is respectfully submitted that, in paragraph [0133], *Nilsson* only discloses that, to determine the exact amount of data in the client's buffer, the server also needs to know the <u>timestamp</u> of the data packet that the client is currently decoding under the assumption that the client's clock does not drift significantly from the server' clock. In paragraph [0134], Nilsson discloses that, in the rare case of <u>a large amount of drift accumulating</u>, the client can <u>warn</u> the server by use of a control message, such as the one described earlier that is sent for <u>decoding buffer underflow</u>.

According to *Nilsson*, in case of <u>decoding buffer underflow</u>, the server is notified of the buffer underflow so that the server will send packets as quickly as possible (paragraph [0126]-[0127]). Furthermore, even if the server knows what the <u>timestamp</u> of the data packet the client is currently decoding, the server does not know the clock drift between the server and the client. While the server may be able to estimate the cumulating data amount in the client from the timestamp of the data packet the client is currently encoding, the server may not be able to know the exact cause of data cumulating in the client. Nevertheless, *Nilsson* fails to disclose providing to the server information indicative of at least one parameter defined in the client, wherein said at least one parameter comprises a shift amount in time indicative of a clock drift between the server and the client.

<u>B.3.1 *Ravi, Wang* and *Nilsson* Fail to Render Claims 6, 7, 10, 16, 25 and 33-34 Obvious</u>

For the above reasons, *Ravi*, in view of *Nilsson*, fails to render independent claim 6 obvious.

For the same reasons, *Ravi*, in view of *Nilsson*, fails to render independent claim 32 and dependent claims 7, 10, 16, 25 and 33-34 obvious

# CONCLUSION

Claims 1-7, 9-17, 19-34 and 36 are allowable. Early allowance of all pending claims is earnestly solicited.

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